INTERPOLATOR

[0001] This application claims priority to German
Patent Application 101 12 275.6, filed March 14, 2001,
the entire contents of which are hereby incorporated by
reference.

BACKGROUND OF THE INVENTION

Field of the Invention

[0002] The invention relates to an interpolator with
fractionally adjustable interpolation instants. For
example, such an interpolator is used in a resampler.

Related Art

[0003] A resampler with a corresponding interpolator
is disclosed in EP 0 665 546 A2. In that case, the
interpolator   comprises   two   FIR   filters   with   a
corresponding   ·coefficient   memory.   What   is
disadvantageous in the known interpolator is that it is
designed in only one stage. If a high accuracy is
demanded, the phase step size of the interpolator must
be relatively small, so that a relatively large number
of coefficients have to be stored in the coefficient
memory, namely a complete set of coefficients for each
phase step size. This leads to a high memory outlay
and, in the case of monolithic integration of the
resampler, to a large chip area for the coefficient
memory.   Furthermore,   the   memory   access   time   is
relatively long, as a result of which the processing
speed is reduced.

SUMMARY OF THE INVENTION

[0004] The present invention is based on an object of
specifying an interpolator and an interpolation method
with which high interpolation accuracy may be realized
in conjunction with a relatively low memory outlay.

[0005] This object, and other objects of the present invention, is achieved by way of the features of claim 1 with regard to the interpolator and by way of the features of claim 6 with regard to the interpolation method. The dependent claims relate to additional advantageous developments of the present invention.

[0006] The present invention is based on the approach of performing an interpolation in the center of the sampling period of the input signal sequence in a half band filter, and thus halving the sampling periods of the input signal sequence. Therefore, given the same accuracy, the number of phase steps of the polyphase filters connected downstream only has to be half as large as without the half band filter connected upstream. The memory outlay for the coefficients of the polyphase filters is already halved by this measure. A further reduction of the memory outlay results from the linear interpolation connected downstream.

[0007] On account of the fact that two polyphase filters are used, one polyphase filter interpolating in the pattern of the predetermined phase steps before the interpolation instant and the other polyphase filter interpolating in the pattern of the phase steps after the interpolation instant, and the further fact that the linear interpolation filter interpolates linearly between the two interpolation results of the two polyphase filters, the accuracy is increased. Moreover, on account of the linear interpolation connected downstream, therefore, it is possible to reduce the phase step width in the polyphase filters, as a result of which the memory outlay is reduced. A further considerable advantage includes the fact that significantly fewer multipliers have to be present in

2

the polyphase filters in order to achieve the same interpolation accuracy.

[0008] In order to limit the bandwidth, a second half band filter may be connected upstream.

[0009] In order not to have to perform an increase in the processing clock rate in the polyphase filters despite the up-sampling in the first half band filter, the even-numbered output values of the first half band filter may be fed to a first series of serially arranged delay elements, while the odd-numbered output values of the first half band filter are fed to a second series of serially arranged delay elements. The multipliers of the polyphase filters are alternately connected to the first series and second series of delay elements via a changeover device (multiplexer).

[0010] If the interpolation instant lies between the last phase step of the polyphase filter and the next sampling instant, then the output signal sequence of the first half band filter shifted by a sampling period must be fed to one of the two polyphase filters. This may be done by way of the changeover device (multiplexer), which in this case accesses the series of delay elements in a manner offset by one delay element.

BRIEF DESCRIPTION OF THE DRAWINGS

[0011] An exemplary embodiment of the present invention is described in more detail below with reference to the drawings, in which:

[0012] Figure 1 illustrates a block diagram of an interpolator according to the present invention;

[0013] Figure 2 illustrates a spectrum of the input signal and of the output signal together with transfer functions of a first half band filter, of the polyphase filter and of a linear interpolation filter;

[0014] Figure 3 illustrates the spectrum of the input signal and of the output signal together with the transfer functions of the second half band filter, of the first half band filter, of the polyphase filter and of the linear interpolation filter;

[0015] Figure 4 illustrates a further block diagram of the interpolator according to the present invention;

[0016] Figure 5 illustrates a diagram for elucidating the interpolation method according to the present invention;

[0017] Figure 6 illustrates the coefficients of the two polyphase filters as a function of the phase step;

[0018] Figure 7 illustrates a block diagram of a first exemplary embodiment of a polyphase filter of the interpolator according to the present invention;

[0019] Figure 8 illustrates a block diagram of a second exemplary embodiment of a polyphase filter of the interpolator according to the present invention; and

[0020] Figure 9 illustrates a diagram for elucidating the linear interpolation.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

[0021] Figure 1 illustrates a block diagram of an interpolator 1 according to the present invention. Via

a changeover device (multiplexer MUX) 2, a digital
input signal sequence x(k) can optionally be passed
directly to a first half band filter (HBF$_2$) 3 or
indirectly, via a second half band filter (HBF$_1$) 4 to
the first half-band filter 3. The second half band
filter 4 has a transfer function H$_1$(f) and does not
change the sampling rate (up-sampling factor up$_1$ = 1).
The first half band filter 3 has a transfer function
H$_2$(f) and increases the sampling rate by the factor 2
(up-sampling factor   up$_2$ = 2). The first half band
filter 3 generates an intermediate signal sequence
z(k).

[0022] A polyphase filter (PPF) 5 is connected
downstream of the first half band filter 3. As can be
seen from figure 4, the polyphase filter 5 is
subdivided into a first polyphase filter 5a and second
polyphase filter 5b, which in each case has the
transfer function H$_3$(f) and increases the sampling rate
by, for example, a factor 128 (up-sampling factor up$_3$ =
128).

[0023] A linear interpolation filter (LINT) 6 having
the transfer function H$_4$(f) is connected downstream of
the polyphase filter 5. The output signal sequence y(k)
is available at the output of the linear interpolation
filter 6.

[0024] Figure 2 illustrates a permissible spectrum
In(f) – with regard to the bandwidth – of the input
signal sequence x(k) together with the transfer
functions H$_2$(f) of the first half band filter 3, H$_3$(f)
of the polyphase filter 5 and H$_4$(f) of the linear
interpolation filter 6. The spectrum Out(f) of the
output signal sequence y(k) is produced by multiplying
the input spectrum In(f) by the transfer functions

$H_2(f)$, $H_3(f)$ and $H_4(f)$. In this case, the changeover device 2 is switched in such a way that the second half band filter 4 is bypassed.

[0025] As is known, the transfer function $H_2(f)$ of the half band filter 3 is symmetrical with regard to $f/f_{r2}=0.25$, the ramp beginning at 0.31 f_{r1} or 0.115 f_{r2}. In this case, f_{r1} is the sampling period of the input signal sequence $x(k)$ and f_{r2} is the sampling period of the intermediate signal sequence $z(k)$ at the output of the first half band filter 3. While the Fourier transform of the input signal sequence $x(k)$ is periodically repeated at the interval f_{r1}, the transfer function $H_2(f)$ is periodically repeated at the interval f_{r2}.

[0026] The transfer function $H_3(f)$ of the polyphase filter 5 has a ramp which likewise begins at $0.31 \cdot f_{r1}$ or $0.115 \cdot f_{r2}$ and is periodic at $f_{r3}=128 \cdot f_{r2}$. Parasitic passbands occur in between, but they lie in spectral ranges in which the transfer function $H_2(f)$ of the first half band filter 3 blocks. The task of the transfer function $H_3(f)$ includes suppressing the repetition spectra of $H_2(f)$.

[0027] The transfer function $H_4(f)$ of the linear interpolation filter 6 falls continuously and reaches a zero point at $f/f_{r3}=1$. Consequently, the transfer function $H_4(f)$ of the linear interpolation filter 6 blocks where the passbands of the transfer functions $H_2(f)$ and $H_3(f)$ coincide at $f/f_{r3}=1$. As long as the bandwidth f_{pass} of the input signal sequence $x(k)$ is smaller than $0.31 \cdot f_{r1}$, the input signal sequence $x(k)$ is mapped onto the output signal sequence $y(k)$ in a manner free from distortion and interference.

[0028] If the input signal sequence x(k) is not band-limited, the second half band filter 4 is to be connected upstream, the transfer function $H_1(f)$ of which is illustrated in figure 3. The transfer function $H_1(f)$ of the second half band filter 4 has a ramp which begins at $0.19 \cdot f_{r1}$ and reaches its zero point at $0.31 \cdot f_{r1}$. Consequently, the second half band filter 4 performs band limiting to the frequency range in which the transfer function $H_2(f)$ of the first half band filter 3 is constant.

[0029] Figure 4 shows a somewhat more detailed block diagram of the interpolator 1 without the second half band filter 4 and without the changeover device 2. It can be seen from figure 4 that a first polyphase filter 5a and a second polyphase filter 5b are connected downstream of the first half band filter 3, both the interpolation value $y_{PPF\_L}(k)$ of the first polyphase filter 5a and the interpolation value $y_{PPF\_R}(k)$ of the second polyphase filter 5b being fed to the linear interpolation filter 6. The interpolation instant $\Delta t/T_{r1}$ relative to the sampling period $T_{r1} = 1/f_{r1}$ of the input signal sequence x(k) is prescribed by a control signal S generated by a timer 7.

[0030] The interpolation method according to the present invention is diagrammatically illustrated in figure 5. An analog signal represented as a function of time is sampled at the instants $k \cdot T_{r1}$. These samples form the input signal sequence x(k). The input signal sequence x(k) is represented by filled-in circles in figure 5. In the example illustrated, the sampling instant is intended to be reduced by the resampler. Consequently, the interval between the sampling instants of the output signal sequence y(k) is larger, as illustrated in figure 5. The task of the

interpolator 1 is to interpolate the amplitude values at the sampling instants of the output signal sequence $y(k)$, which are represented as rhombi in figure 5, from the input signal sequence $x(k)$. The interpolation instant $\Delta t/T_{r1}$ in relation to the preceding sampling instant of the input signal sequence $x(k)$ is transferred by the timer 7 to the interpolator 1 by way of the control signal S. Furthermore, an enable signal E is transferred, which indicates whether or not a value of the output signal sequence $y(k)$ is to be interpolated in the respective sampling interval of the input signal sequence $x(k)$.

[0031] During the interpolation, the procedure according to the present invention is such that, in the first half band filter 3, an interpolation is in each case performed in the center of each sampling period T_{r1} of the input signal sequence $x(k)$ and an intermediate signal sequence $z(k)$ is thus generated. Since the interpolation is always effected in the center of the sampling period T_{r1}, that is to say with an always identical phase shift in relation to the input signal sequence $x(k)$, the complexity in the half band filter 3 is limited insofar as the half band filter 3 can operate with fixed coefficients. Given the same phase step size T_{r3}, however, the number of phase steps in relation to the sampling period T_{r2} of the intermediate signal sequence $z(k)$ is only half as large as in relation to the sampling period T_{r1} of the input signal sequence $x(k)$. Consequently, only half as many coefficients have to be kept ready in the coefficient memory of the polyphase filters 5a and 5b, which means that the memory outlay is halved for the same accuracy. Moreover, significantly fewer multipliers are required in the polyphase filters 5a and 5b.

8

[0032] The first polyphase filter 5a interpolates the intermediate signal sequence z(k) at an instant t_L which lies in the predetermined pattern of possible interpolation instants of the polyphase filters before the interpolation instant Δt prescribed by the control signal S. By contrast, the second polyphase filter 5b interpolates the intermediate signal sequence z(k) at an instant t_R which lies in the predetermined pattern of possible interpolation instants after the interpolation instant Δt prescribed by the control signal S.

[0033] Finally, a linear interpolation between the two interpolation values y_{PPF_L} and y_{PPF_R} found at the instant t_L and t_R is effected in the linear interpolation filter 6. The final interpolation value y_{LINT} is found by way of the subsequent linear interpolation.

[0034] Figure 7 shows an exemplary embodiment of the polyphase filter 5a, which is identical to the polyphase filter 5b. A series of delay elements (registers) 8a, 8b, 8c and 8d can be seen. A greatly simplified example with only four delay elements 8a-8d is illustrated in this case. The delay elements 8a-8d are respectively connected via a changeover device (multiplexer MUX) 9a, 9b, 9c and 9d to a respective first input 10a, 10b, 10c and 10d of a multiplier 11a, 11b, 11c and 11d, respectively. The second input 12a, 12b, 12c and 12d of the multiplier 11a, 11b, 11c and 11d, respectively, is connected to the coefficient memory 13a, 13b, 13c and 13d, respectively. The control signal $)t_B/T_{r2}$ prescribes the phase step for which the polyphase filter 5a is to interpolate.

[0035] In the greatly simplified exemplary embodiment illustrated in figures 5 and 7, the sampling period T_{r2}

9

of the intermediate signal sequence $z(k)$ is subdivided
into eight phase steps, i.e. the polyphase filter 5a
can interpolate in a pattern of eight possible
interpolation instants. The phase step number can
therefore be represented by three bits, as is
illustrated on the far left in figure 6. For each of
the eight possible phase steps, an associated
coefficient is in each case selected from the
coefficient memories 13a-13d. By way of example, in the
case of the phase step 001, the coefficient a_2 is
selected for the multiplier 11a, the coefficient b_2 is
selected for the multiplier 11b, the coefficient c_2 is
selected for the multiplier 12c and the coefficient d_2
is selected for the multiplier 12d. In the case of the
last and seventh phase step 111, the set of
coefficients a_8, b_8, c_8 and d_8 is selected. The two
polyphase filters 5a and 5b in each case interpolate at
adjacent interpolation instants, as can be seen from
figure 5.

[0036] A special feature occurs if the polyphase
filter 5a interpolates at the last phase step of a
sampling interval of the intermediate signal sequence
$z(k)$, that is to say at the seventh phase step 111 in
the example, but the polyphase filter 5b already
interpolates at the first phase step 000 of the
succeeding next sampling interval of the intermediate
signal sequence $z(k)$. In the exemplary embodiment
illustrated in figure 7, this problem is solved in that
the coefficient 0 for the first phase step 000, for
which there is no phase shift relative to the last
sample of the intermediate signal sequence, is in each
case read from the coefficient memories 13a-13d.
However, the next sample $z(k+1)$ of the intermediate
signal sequence is accessed via the changeover devices
9a-9d. It can be seen from figure 6 that the

10

coefficients of the polyphase filter 5b are cyclically shifted in each case by one phase step relative to the coefficients of the polyphase filter 5a, i.e. if the polyphase filter 5a is allocated the set of coefficients a_8, b_8, c_8, and d_8, the polyphase filter 5b is already allocated the set of coefficients 0,1,0,0.

[0037] The outputs of the multipliers 11a-11d are fed to a summer 14 in a customary manner. The output 15 of the summer 14 is connected to the linear interpolation filter 6.

[0038] Figure 8 shows a second exemplary embodiment of the polyphase filters 5a and 5b, respectively. The sampling rate f_{r2} of the intermediate signal sequence $z(k)$ is twice as large as the sampling rate f_{r1} of the input signal sequence $x(k)$. The exemplary embodiment of the polyphase filters 5a and 5b which is illustrated in figure 7 would therefore have to be operated at twice the clock rate. It is advantageous, however, to use a uniform clock rate in the interpolator 1. In order to achieve this, in the exemplary embodiment illustrated in figure 8, the delay elements 8a, 8b, 8c... are arranged in two series 16 and 17, respectively. The odd-numbered output values of the first half band filter 3, that is to say the odd-numbered values $z(2k+1)$ of the intermediate signal sequence, are fed to the first series 17 of delay elements 8a, 8c... By contrast, the even-numbered output values of the first half band filter 3, that is to say the even-numbered values $z(2k)$ of the intermediate signal sequence, are fed to the second series 16 of delay elements 8b... The two series 16 and 17 of delay elements 8a, 8b, 8c can then be clocked with the clock f_{r1}. Since the values of the intermediate signal sequence $z(k)$ are in each case allocated alternately to the two series 16 and 17, the

connection allocation of the changeover elements 9a-9d is to be modified in the manner illustrated in figure 8. In each case one input of the changeover devices 9a-9d is connected to the first series 17 and the other input of the changeover devices 9a-9d is connected to the second series 16.

[0039] Figure 9 shows the procedure in the linear interpolation filter 6. The polyphase filter 5a generates the interpolation value $y_{PPP\_L}(k)$. The polyphase filter 5b generates the interpolation value $y_{PPP\_R}(k)$. The polyphase filter 5b generates the interpolation value $y_{LINT}(k)$. The linear interpolation filter 6 interpolates in accordance with the time offset $\Delta t_c/T_{r3}$ that still remains (cf. figures 5 and 9), as a result of which the final interpolation value $y_{LINT}(k)$ is obtained.

[0040] In the interpretation of the control signal S, the procedure may be such that the most significant bit decides whether the interpolation is performed in the first half or in the second half of the sampling interval T_{r1}, a series of medium significant bits define the phase step of the polyphase filters 5a and 5b and the least significant bits not yet taken into account drive the linear interpolation filter 6.

[0041] The present invention is not restricted to the exemplary embodiment illustrated and can also be used e.g. in other embodiments of the polyphase filters 5a and 5b.